multichannel version of MPEG Layer II at 640 kbit/s with the following three multichannel versions of MPEG AAC (then labeled NBC): main profile at 320 kbit/s, main profile at 256 kbit/s and low complexity profile at 320 kbit/s. The tests were performed at two venues in parallel (the BBC and NHK) using the methods described in ITU-Recommendation BS.1116. These methods are designed to estimate the worst case performance of codecs. The main test results are shown in Figure 13. This figure shows the overall mean rating (across all 10 audio test materials and all listeners (23 at the BBC and 16 at NHK)) for each of the codecs tested.

The two AAC codecs tested at 320 kbit/s, namely the main profile and the low complexity profile, met the ITU-R broadcast quality requirement: none of their 10 audio test materials had mean diffgrades lower than –1.0. The low complexity profile was found to be only marginally lower in quality than the main profile. For the AAC main profile at 256 kbit/s, although the overall mean grade is above –1.0 in Figure 13, several audio materials had mean diffgrades below –1.0.
These tests were performed in 1996 on a very early version of MPEG AAC. Since then, progress has been made and it is reasonable to assume that state-of-the-art implementation of AAC would yield, today, audio broadcast quality at bit rates smaller than 320 kbit/s for 5.1 channels, say in the range 256 kbit/s to 320 kbit/s.

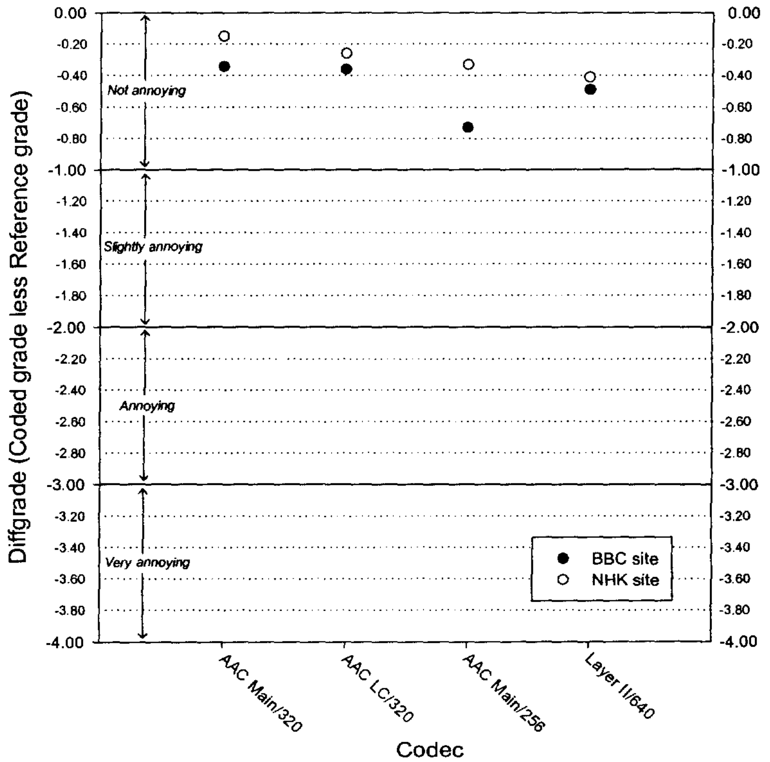

Figure 13: Comparison of overall mean opinion scores between multichannel audio codecs [a]

5.2 Error sensitivity and extra error correction overhead

Although the MPEG Layer II in its two-channel mode was found to be relatively robust to channel errors and requires typically BER of between 10^{-3} to 10^{-4} for threshold of audibility of audio impairments (see section 4.2.4), early indications are that new and more efficient multichannel audio systems will be more susceptible to channel errors due to the bit allocation schemes that they used. It is expected that these bit streams will have to be protected to a minimum level of BER of 10^{-5}. This means that a channel coding made of a concatenation of a outer block code and a inner convolutional code will need to be implemented to provide the required protection at minimum Eb/No.

6. System data multiplex structure

6.1 Stream mode versus packet mode

Data services differ in their requirements, most notably when it comes to bandwidth. Some services, such as streaming audio or video, generate data at a constant rate while they are active and are best served by a "virtual circuit" which provides constant bandwidth. Other services, such as file transfers and e-mail type messaging, are much more intermittent in their activity and bandwidth requirements. Even during their periods of intermittent activity, these latter services typically do not have fixed bandwidth requirements; instead, the requirement could be characterized as a "maximum acceptable transfer time". The Internet, being packet-switched, is well suited for this latter type of service. However, it cannot provide constant bandwidth virtual circuits, and it is necessary to resort to buffering and other tricks to obtain acceptable performance with streaming services (and as many Internet users can attest, even those measures do not always suffice).

Packetizing data that does not really need such treatment is also inefficient, since it introduces unnecessary overhead. On the other hand, in a circuit-switched network, it is inefficient to dedicate a constant bandwidth circuit to intermittent packet-type services that do not keep the "pipe" filled. Efficiency is of particular concern in wireless data services, since the radio spectrum is a limited and precious resource. A datacasting system should therefore offer both stream and packet-type services. Fortunately, this is relatively easy to do in a radio broadcast scenario, since there are no multiple hops and routers involved.

6.2 Eureka-147 DAB Multiplex structure

The system multiplex structure of the Eureka-147 DAB system is based on a main service multiplex and a Fast Information Channel (FIC) which is used to signal the multiplex configuration to the receiver. The FIC channel also carries information that is to be accessible by all the receivers tuned to the multiplex whatever the service that is being decoded (alert warning, etc.) This FIC is heavily error corrected and is not time interleaved so that the capture at the receiver is faster.

The Eureka-147 DAB service multiplex is based on 64 sub-channels of capacity varying from 8 kbit/s to about 1 Mbit/s. Each sub-channel can control its level of error protection (1/2 to 7/8), its scrambling and can carry a data stream such as MPEG-2 Layer-2 coded audio or a packet stream of up to 1024 different packet services. The audio coding can use unequal error protection by varying the protection level according to the sensitivity of specific bits to channel error. The multiplex can be reconfigured dynamically (i.e., 'on-the-fly') at any start of a frame (24 msec) by sending a flag 6 seconds in advance of the change to initiate the process in the receiver. The multiplex can be dynamically reconfigured in a fashion transparent to the user.

6.3 Merits and constraints of MPEG-2 multiplex structure

MPEG-2 transport layer might be a very good candidate to be used in the proposed MMBS system to multiplex the audio, video, and data services into one Transport Stream (TS). The MPEG-2 transport layer was designed for multimedia service applications. It uses packetized data transmission to multiplex all types of data into 188-byte packets. The MPEG-2 transport layer has been widely adopted world-wide for digital television services (satellite DTH, Cable DTV, terrestrial DTV, MMDS and LMDS services) and Digital Versatile Disk (DVD). Using MPEG-2 would allow the MMBS system to easily exchange program and data with other broadcast services at the transport layer. MPEG-2 packets can also encapsulate ATM cells as defined by DAVIC standard and this would allow interfacing with the communication networks. However, it might be difficult for the 188-byte MPEG-2 packets to be carried efficiently by the 53-byte ATM cells over high speed switching networks.

At the moment, ATSC is completing a Data Broadcasting standard, which allows the use of MPEG-2 transport stream to transmit IP and other various data traffic. The final approval of the ATSC data broadcasting standard is expected in October 1999.

The main drawback of using the MPEG-2 transport layer for MMBS is its relatively large, 188-byte, packet size. It was optimized for maximum data throughput with minimum overhead having in mind high capacity services such as video services. In dealing with the relatively small capacity services to be carried by MMBS (audio and data services), there may be a lack of granularity with the MPEG-2 transport which would result in a loss of flexibility and efficiency in carrying the small packet content.

6.4 Suggested multiplex structure

With the current knowledge of communication systems and the pace at which it evolves, the best, most flexible and most future proof approach is to structure the multiplex in independent layers as defined in the OSI reference model. This would ensure that the system can evolve in time and that future services can be accommodated. Smaller packet size than the ones used in MPEG-2 transport layer may be preferred to maximize the system flexibility in carrying small capacity services.

More work needs to be done on this aspect to make sure that the system has maximum flexibility while still allowing fast acquisition by the receiver and allowing special features such as conditional access and software download to upgrade dynamically the receiver functions. By the time the service could be launched (i.e., 2006), software receivers should be the norm and the system should allow for reliable and secure software updates (see section 8).

6.5 Error sensitivity of data services

Data services will require rather high channel error protection since, in a broadcast environment, there is no possibility of identifying the packets to be re-sent because of error. The other means would be to repeatedly send the data packets so that the receiver can capture a final reliable version. This would be a rather wasteful way of providing the service. There is a need, as in the multichannel audio case, for a se of concatenated block and convolutional codes to provide BER protection in the range 10-6 and above for the data services. A common set of codes could be used for both multichannel audio and data. Nevertheless, the multiplex structure may need to allow for some flexibility on this aspect and provide the services that are less sensitive to channel errors with the more basic convolutional coding to preserve capacity.

7. Spectrum capacity

7.1 Expected number of channels in the upper UHF

This system investigation is based on the announcement by the FCC (Docket No. 99-168 [1]) that the current TV-UHF channels 60, 61, 62 and 65, 66 and 67 would become available for potential spectrum auctioning for commercial purposes after January 1, 2001. This spectrum would become available in 2006. The two actual frequency ranges are 746-764 MHz and 776-794 MHz with the median frequency being 770 MHz. Figure 14 presents an example of MMBS channelization with 4 blocks of 1.5 MHz occupying each of the six 6-MHz UHF channels.

7.2 Spectrum efficiency

A series of laboratory tests were conducted in 1994/95 by the Digital Audio Radio (DAR) Subcommittee of the Electronic Industries Association and the DAB Subcommittee of the National Radio System Committee [26]. These tests evaluated the performance of nine DAR systems in simulated mobile channels representative of four different geographical environments (urban slow, urban fast, rural and obstructed terrain). Details of the test parameters can be found in [27].

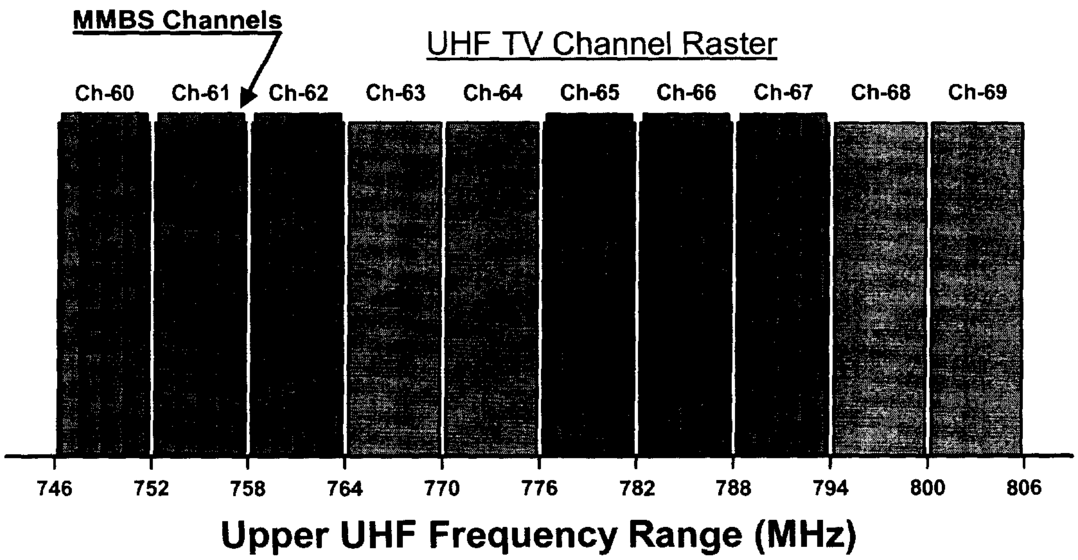

Figure 14: An example of an MMBS channellization in the upper UHF

Figure 15 presents the spectral vs power efficiency of the four DAR systems which were successfully tested in the mobile multipath channels. The spectral efficiency represents the number of bit/s of useful information (in this case audio) that can be transmitted per unit of RF signal bandwidth (Hz), while the power efficiency represents the ratio of energy per useful information bit over the noise power spectral density (E_b/N_o) in dB. This Figure shows, as references, the theoretical capacity of both the AWGN and Rayleigh (one branch diversity) channels. The results were presented, for each of the four systems, in the form of an error bar covering the range of E_b/N_o (from minimum to maximum values) at the threshold of audibility of errors obtained in the mobile multipath channels. Of the systems successfully tested in three out of the four mobile multipath channels, the two implementations of the Eureka-147 DAB system (224 kbit/s and 196 kbit/s audio channels) are the most power efficient systems (with E_b/N_o values ranging from 16.2 to 20.4 dB for Version 1 and from 17.3 to 21 dB for Version 2) but the least bandwidth efficient ones (0.75 bit/s/Hz). The AT&T system is less power efficient (E_b/N_o ranging from 21.5 to 30.1 dB) but, at 0.85 bit/s/Hz, it is 13%

more spectrally efficient that the two versions of the Eureka-147 DAB system. The AT&T/Amati system is the most spectrally efficient of the four systems at 1.09 bit/s/Hz, but it is less power efficient than these two systems and has failed on two of the four mobile channels.

The above results give an indication of the power and spectral efficiency of digital broadcast systems in the mobile environment. The most robust systems tested achieved a spectral efficiency in the range 0.75-0.85 bit/s/Hz. The system with a spectral efficiency of 0.85 bit/s/Hz required more power than the systems at 0.75 bit/s/Hz, showing the trade-off that can be made between power and spectral efficiency.

The power efficiency of the Eureka-147 DAB system can easily be deduced from the fact that DQPSK modulation is used (i.e., 2 bit/s/Hz); ½ code rate is used to protect against errors (resulting in 1 bit/s/Hz); then a 20% guard interval is used to buffer out the inter-symbol interference, thus providing 0.8 bit/s/Hz. In addition, the overhead arising from the need of the null symbol, the synchronization symbol, the Fast Information Channel data to signal the structure of the multiplex, this can easily account for the equivalent of some 5 % of the total capacity. One arrives at a final spectral efficiency of 0.75 bit/s/Hz.

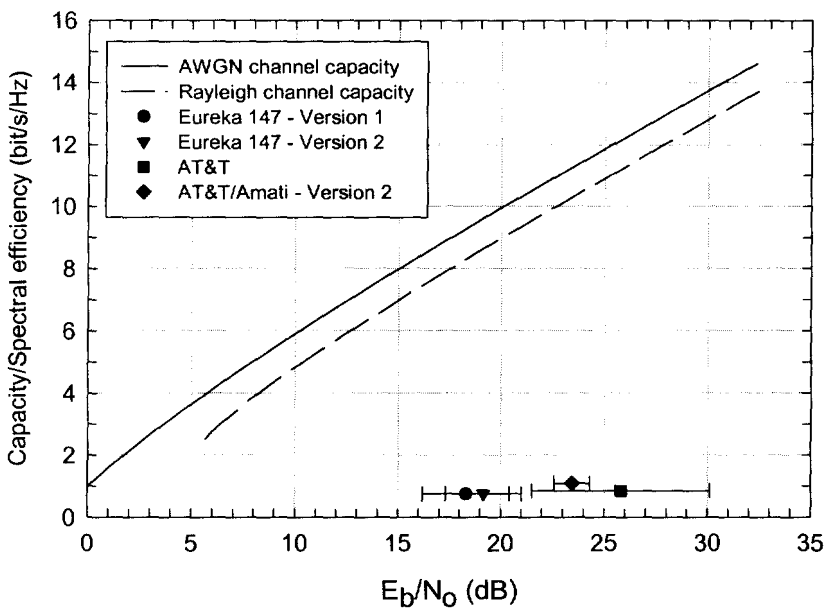

Figure 15: Spectral vs power efficiency of DAR systems in mobile multipath channels

Although a slightly more efficient means of synchronizing the receiver and signaling the structure of the multiplex could be used, this advantage would be counter-balanced by the need for some guard bands between each 1.5 MHz channels to facilitate the receiver tuning and frequency capture of the channel and to reduce adjacent channel interference. A reasonable range is probably between 0.7 and 0.85 bit/s/Hz, depending on the data overhead and mainly the extent of the guard interval, trading-off capacity for system flexibility. For the purpose of the MMBS system strawman, a value of 0.75 bit/s/Hz will be used.

7.3 Total spectrum capacity

The total spectrum that the FCC is expected to auction for this type of broadcast service is six 6-MHz TV channels, which amounts to a total of 36 MHz. Using the reference spectral efficiency suggested in the previous section, this amounts to a total useful data capacity of some 36 x 0.75= 27 Mbit/s.

Assuming that the bit rate required for a very high quality 5.1 multichannel audio program is 288 kbit/s (middle of the range established in section 5.2) and adding a 64 kbit/s ancillary data channel accompanying each multichannel audio program plus 8% overhead for the necessary extra forward error coding due to the sensitivity of these services to channel errors (e.g., RS(204,188) [28]), this results in a total bit rate of some 375 kbit/s per complete multichannel audio/data program (including overhead).

Consequently, the total 36 MHz RF spectrum could be used to provide up to 72 independent multichannel audio /data programs. Because of the limit on minimum bandwidth of 1.5 MHz to take advantage of the frequency diversity needed in a frequency selective multipath channel, this spectrum could be arranged in 24 multiplexed "ensembles" occupying 1.5 MHz RF channels which can carry three multichannel audio/data programs each.

7.4 Practical spectrum capacity

7.4.1 Co-channel interference consideration

Although the total spectrum of 36 MHz would allow for some 72 multichannel audio/data programs in 24 RF channels of 3 programs, all this capacity would not be available in a same area due to the need for spectrum re-use in neighboring areas. A simple approach at spectrum re-use is to start with a regular lattice of hexagonal service areas, as is often used in planning the cellular mobile radio-telephone service. The frequency repetition number in this case is 7 if the same frequency can be used outside the first layer of adjacent service areas. This would translate into a rather small spectrum availability for each area, that is 24 channels divided by 7 resulting in 3 RF channels per service area. In reality, better spectrum efficiency is likely to be achieved.

The DTV allotment plan for the USA was used as a reference to obtain a first assessment of the service capacity for MMBS in various cities. The DTV plan is based on a service availability of F(50,90) for the wanted signal and F(50,10) for the interfering signal. The DTV into DTV co-channel protection ratio is 15.3 dB. This resulted in co-channel separation distances (transmitter-to-transmitter) of 196 km (Zone 1, as defined by the FCC) and 224 km (Zones 2 and 3) with typical coverage area radius of 80 km. [29] The typical co-channel separation distances, edge-to-edge, are therefore 36 and 64 km.

This results in a rather tight plan which allows rather high frequency re-use. Table 5 summarizes the situation in seven main US cities, in terms of the interim NTSC and DTV stations as well as an estimate of the total DTV stations that could be brought on-the-air after the closing of the NTSC stations, assuming that all the frequencies in the core spectrum (Ch. 2 to 51 except 37) can be re-used in the same area by new DTV stations. [30] As can be seen, the frequency repetition number based on a total of 49 available channels is much less than 7. As an example, if the DTV plan was used as a model for MMBS channel allocation, Washington, DC with a frequency repetition number of 3 would be allowed 24 /3 = 8 MMBS channels therefore 24 multichannel audio/data programs.

Main US Cities	Current NTSC Stations	Interim DTV Stations	Total DTV Stations	Frequency repetition number
New York	7	7	14	3.77
Los Angeles	11	11	18	2.88
San Francisco	10	10	20	2.58
San Diego	6	6	12	4.45
Sacramento	6	6	12	5.44
Denver, CO	10	10	20	2.58
Washington, DC	8	8	16	3.06

Table 5: Summary of the NTSC and DTV stations assigned to 7 markets in the interim and final steps of the DTV implementation

Unfortunately, the DTV plan cannot be used directly in this case because there is a number of basic differences in the required planning assumptions. Although the required co-channel protection ratio is relatively similar (15.3 dB for DTV in a Gaussian channel versus 12.2 dB[1] for COFDM/DQPSK in a Rayleigh channel), the assumed receiving installation is very different (directional antenna at 10 m for ATV versus omni-directional antenna at 1.5 m for MMBS), and the service availability criterion is very different (F(50,90) for DTV versus a suffested F(90, 90) for MMBS). In fact, the availability objective for the MMBS service is to be developed very carefully because this service is aimed at providing a very reliable service to the mobile market. Being for mobile, the location variability translates into time variability and an overall time availability between 90% and 99% may need to be precisely defined.

In any event, this is likely to translate into a requirement of a much higher wanted-to-unwanted signal differential at the edge of the service area, resulting in a much larger separation distance than in the DTV case. However, a new approach in planning a broadcast service was used to maximize the spectrum re-use in the case of the DAB service in Europe during the Wiesbaden Planning Conference [31] and in Canada [32] based on the assumption that networks of on-channel transmitters will be used to cover the service areas. In the Wiesbaden plan, this translated into the assumption that most of the interference at the edge of the service area would be produced by the closest repeater located only 17 km from the edge of the service area rather than a much higher power transmitter located at the center of the service area. Obviously, the rate at which the interfering power drops off at the edge is much faster in the case of a low power repeater at 17 km that from a high power transmitter located at 80 km from the edge.

This concept allowed the Wiesbaden plan to be based on a separation distance (edge-to-edge) of some 61 km at 1.5 GHz. Because of the fact that propagation at 770 MHz will allow the signal to reach

[1] In developing the link budget in section 4.2.7 to arrive at the minimum field strength requirement, 2 dB was set aside for interference consideration. As it was proven in the case of the Eureka-147 DAB system, the effect of co-channel interference on the error performance can be assimilated to that of noise (i.e., co-channel protection ratio is equivalent to the C/N requirement), this translates into a requirement of 26.2 dB difference between the medians of the wanted and unwanted signals to meet the 2 dB noise equivalent apportionment set aside in the link budget assuming a 5.5 dB standard deviation for the propagation effects (this value is becoming broadly accepted for wideband digital broadcast services. [31]). This also corresponds to an equivalent protection ratio of 12.2 dB to be used between the 90% location availability contour for the wanted signal and the 10 % location availability contour for the interfering signal (26.2 –1.28x5.5 –1.28x5.5 = 12.2).

over a longer distance, 61 km may be somewhat of an optimistic value. However, considering that the Wiesbaden plan was based on 99% service availability rather than 90%, this would most likely more than compensate for the increased signal propagation at lower frequency. The range 36-64 km as used in the DTV plan may not be that far from the right value to be used for MMBS. Further work is clearly needed to assess the impact of the omni-directional receive antenna assumed for MMBS in this context.

As a first attempt, it is proposed to use the DTV frequency re-use factors as an indication of the potential spectrum capacity of the MMBS service. Table 6 summarizes the capacity expected for 7 main cities in the US. It is worth remembering that this is based on the use of networks of on-channel repeaters. If single high power transmitters were to be use instead, much lower capacity (probably by a factor of ½ or less) would be available.

Main US Cities	Frequency repetition number	MMBS frequency blocks per city	Total multichannel audio/data programs per city
New York	3.77	6	18
Los Angeles	2.88	8	24
San Francisco	2.58	9	27
San Diego	4.45	5	15
Sacramento	5.44	4	12
Denver, CO	2.58	9	27
Washington, DC	3.06	8	24

Table 6: Summary of the expected MMBS capacity in seven major US cities

7.4.2 Adjacent channel consideration

Adjacent channel interference is a problem that is found in the vicinity of an interfering transmitter, while one wants to receive from a distant transmitter. The differential in signal power may be large so that the receiver, even if it is operated in its linear range, has difficulty in rejecting the adjacent channel. If COFDM is used, the FFT used in the receiver has an inherently high selectivity against carriers that are outside the range to be considered. Furthermore, guard bands will need to be set aside between each 1.5 MHz RF channel to ease the receiver frequency tuning and coarse tracking. This should also reduce to some extent the adjacent channel interference. It was found, with the Eureka-147 DAB receivers, that adjacent channel rejection is only a few dB less than the rejection of any other channel in the band, the latter being caused by non-linear operation of the receiver front-end (see section 7.4.3).

7.4.3 Receiver front-end saturation consideration

In the case of a receiver located in the close vicinity of an interfering transmitter, the RF front-end may be driven into saturation and create a high level of intermodulation. The strategy in the receiver for the AGC is based on a trade-off between receiver sensitivity and non-linearity. The non-linearity is usually due to the saturation of the first amplification stages and results in intermodulation products falling within the channel bandwidth, therefore not removed by the IF filter. Typical maximum receiver input levels of –15 dBm should limit this problem to a very local area around a transmitter. Furthermore, at 770 MHz, it is reasonable to think that antenna manufacturers will be able to control the vertical pattern of the transmit antenna so that no excessive levels of signal field strength is directed towards the ground. Most receiver non-linearity should be avoided by this relatively simple measure.

7.4.4 Advantage of co-located transmitters

Co-location of transmitters is a very effective method of avoiding large signal differentials between transmissions in a given area. Since all signals are transmitted from the same tower, their relative power is preserved throughout the service area, therefore reducing the possibility of adjacent channel interference and receiver non-linearity operation. Very simple measures as this one would permit the use of the spectrum capacity to be limited only by co-channel interference considerations. This is even more important in the case of networks of on-channel repeaters where the multiple transmit installations could be shared to transmit many RF channels, leading to an overall cost reduction in the infrastructure implementation.

8. Receiver model

By the time the MMBS service is to become available to the public (i.e., 2006), a different and more flexible kind of receivers is likely to be produced by the manufacturers. These receivers are beginning to appear on the market even today. "Software receivers" as they are called will be easily programmable for handling various types of services and even for changing the demodulation technique to adapt to other transmission formats. They could even become upgradable through software download to keep track of the latest developments in technology, such as the latest and most efficient audio codecs and datacasting services. The main differences between these receivers will be the local resources that they have in terms of interface to the user and the computing power available. Software downloads will ensure that there is a link between the services delivered through the channel and the local resources of the receiver. Services will therefore be allowed to evolve as long as the simplest receivers can make sense out the incoming bit stream and can present it to the user through its limited local resources (e.g., multichannel sound and multimedia services being handled by a simple monophonic audio receiver).

In this context, there is a need to focus one's attention on the lower layers of the system at an early stage in system development. In particular, the physical layer needs to be fully defined since it will be used to transmit the software upgrades for the different receivers. Upper layers need only to be defined in the future and may in fact be upgraded in time by software download. In fact, the DSP power of this receiver could be used to demodulate various modulations such as AM and FM as long as the RF front-end resources exist in the receiver to bring the signal to a low IF or baseband for analog to digital conversion.

Figure 9 presents a block diagram of an MMBS receiver from a generic point of view. All the blocks inside the DSP box can be modified by software. The main elements differentiating various receivers will be the resources available for information rendition and the corresponding API's. Thus, these receivers will in reality be flexible and upgradable multimedia receivers.

9. MMBS coverage considerations

The San Francisco Bay area was chosen to demonstrate the coverage capabilities of MMBS at 770 MHz because of its demanding features. The presence of heavily built-up areas, numerous hills, water and distant mountains makes it a challenging environment for mobile reception. This same area was selected and used in 95/96 by the EIA-DAR subcommittee to field test the different Digital

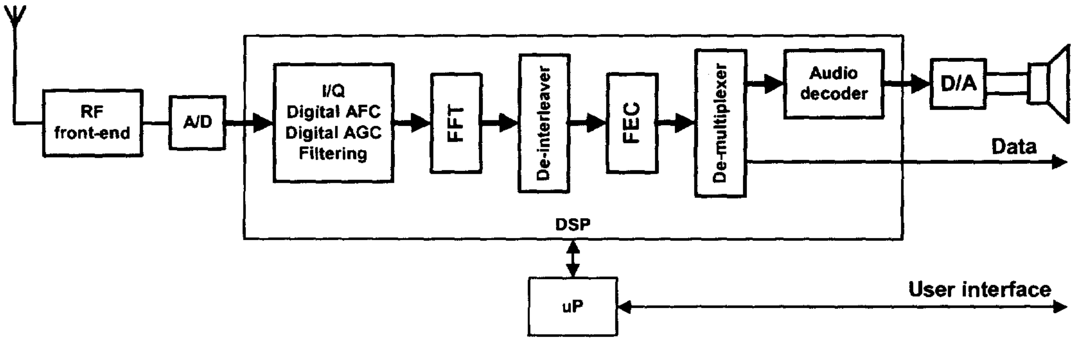

Figure 9: Block diagram of a typical software receiver

Audio Radio systems proposed at the time. These tests proved that the area was indeed very challenging in terms of reception in harsh multipath environment. It also proved that COFDM was very robust in such environment and allowed the successful demonstration of a single frequency network operating at 1.5 GHz. [33] The field tests also proved that the coverage prediction exercise that CRC had performed to optimize the multi-transmitter locations, antenna directivity and power requirements, was very consistent with what was measured afterward in the field.

On this basis, new coverage prediction exercises were carried out at 770 MHz with the technical parameters developed for the MMBS strawman in section 4.2. As compared to the original exercises at 1.5 GHz, the coverage was bound to improve for two reasons: increased diffraction of the transmitted signal around obstacles at 770 MHz which results in higher field strength in shadowed areas; and the increased flexibility of locating on-channel repeaters at larger distances, due to the larger guard interval afforded by the use of a lower carrier frequency, as was covered above in section 4.2.7 in the treatment of the Doppler spread.

9.1 Single MMBS transmitter case

Figure 10 presents the coverage resulting from a single transmitter on Mount San-Bruno using 1 kW EIRP and an omni-directional antenna. The antenna is located at 350m above sea level. A large number of holes appears in the coverage area and would make the service unacceptable . The conventional way to improve the coverage is to increase the transmit power. Figure 11 depicts the coverage produced by the same transmitter except that the EIRP had to be increased to 50 kW to produce an acceptable coverage.

9.2 Multiple on-channel MMBS transmitters case

Bringing the EIRP of Mount San Bruno back to 1 kW, a second transmitter was put on Mount Beacon with the same EIRP and an omni-directional antenna. The antenna height is at 307m above sea level. Figure 13 gives the coverage resulting from this two-transmitter arrangement. The situation is much improved compared to Figure 10 but still needs some help in the southern part. It should be noted that the coverage predictions use a constant 10 dB factor for land cover and local obstructions. In fact this factor varies greatly over the area. The predictions will not represent the great improvement in the San Francisco city core due to the fact that, in the case of the two transmitters, a major network gain will appear since the city core is fed from both sides. This was measured in practice in 1995 at 1.5 GHz during the EIA-DAR field test program.

As a final step in the exercise, a third transmitter was added south of Mount San Bruno to improve the coverage toward San Jose. An exercise of minimizing the total transmit power was also undertaken while still preserving the coverage to the extent possible. This resulted in the Mount San Bruno transmitter requiring 500 W EIRP and the other two transmitters requiring 200 W EIRP. Figure 14 shows the resulting coverage. A summary of the parameters used for the three transmitters can be found in Table 7.

Parameter / Site	**Mount Beacon**	**Mount San Bruno**	**South Transmitter**
Latitude	37°51'3" North	37°41'15" North	37°28'29.4" North
Longitude	122°29'51" West	122°26'4" West	122°21'26.5" West
Height above ground level	50m	50m	50m
Height above sea level	256.82 m	300.14	425.97
EIRP	200 Watt	500 Watt	200 Watt
Antenna type	Omnidirectional	Omnidirectional	Omnidirectional

Table 7 : Parameters of the optimized MMBS Single Frequency Network

This example shows the importance of having a large guard interval to work with. The guard interval retained in the MMBS strawman is 125 μsec. This corresponds to a maximum transmitter spacing of 50 km when synchronously fed. The two extreme transmitters (Mount Beacon and South) are located at some 44 km, which is close to this upper limit of 50 km to avoid the appearance of destructive echoes at the receivers. In fact, because of the likelihood of passive echoes coming from the surrounding mountains, this buffer of 6 km will most likely be taken up in many instances.

10. Comments on existing and proposed systems

10.1 FM-IBOC

Following the first round of laboratory and field tests in the early 90's through the EIA/NRSC DAR evaluation process [33], the various proponents of AM and FM in-band on-channel systems have reviewed their design, learning from the extensive set of laboratory test results produced by the EIA/NRSC test program. Although the details of the IBOC systems currently under development are not yet publicly known, indications are that they will all use some sort of multi-carrier modulation, and more precisely some form of OFDM with the carriers orthogonal to each other. One can also expect the presence of a guard interval, plus time and frequency interleaving coupled with forward error correction, thus resulting in different variants of COFDM. The wide use of COFDM is a testimony to its capabilities for broadcasting in a harsh multipath environment. Although the performance against echoes with medium and large excess delays is likely to be similar to what is described here (or better, through more complex FEC techniques which would generate a lower but more abrupt failure characteristic), the performance against short echoes will be poorer due to the much narrower channel bandwidth, which is likely to be less than 100 kHz (see section 3.3).

Although these systems would allow for the use of on-channel repeaters because of the use of COFDM with guard interval, they will unfortunately not be able to take advantage of it until the host FM transmissions are turned off, since on-channel repeaters would impact the FM reception in their neighborhood, due to the large signal differential.

10.2 Eureka-147 DAB

The transmission scheme for the Eureka-147 DAB system was developed some ten years ago for high quality digital audio broadcasting. The system was designed to provide for vehicular, portable and fixed reception using low gain omni-directional receive antennas typically located at 1.5 m above ground. The system is still being augmented to provide for more sophisticated program related datacasting, such as dynamic label service and slide shows, as well as non-program related data services such as traffic information, etc. Further related standards are being developed in Europe for higher layer applications, such as the Multimedia Object Transport (MOT) protocol (ETSI standard EN 301 234).

The Eureka-147 DAB system was especially developed to offer improved performance in multipath and shadowing environments which are typical of urban reception conditions. It is capable of offering various levels of sound quality, up to high quality sound comparable to that obtained from consumer digital recorded media. It uses the MPEG Layer II audio coding. It can also offer various data services and different levels of conditional access, plus the capability of dynamically re-arranging the various services contained in the multiplex. A standard has been developed for the Radio Data Interface (RDI) to be implemented on most receivers to allow data transfer to computers.

Over the last decade, time has proven that the transmission scheme was and still is today a first class system, proven by extensive field testing and acknowledged by the large number of other broadcasting systems adopting the COFDM approach. On the audio source coding side, it has slipped a bit behind the state of the art, as there has been continued improvement in source coding schemes which can provide equivalent audio quality at lower bit rates. The system has been upgraded to provide a compatible extension to multichannel audio transmission using the MPEG II coding.

The Eureka-147 DAB system is being implemented in Europe and in Canada and is being considered by many countries in the world for the next generation of Radio Broadcasting [www.worlddab.org].

10.3 ISDB-T

The family of Terrestrial Integrated Services Digital Broadcasting (ISDB-T) systems has been developed by NHK in Japan with the broad approach of re-farming the current UHF TV band into digital radio and TV [28]. This concept is based on the use of all the features of COFDM with a range of guard intervals (1/4, 1/8, 1/16, 1/32 Tu), a range of convolutional code rates (from 1/2 to 7/8), time and frequency interleaving and the concatenation of a Read-Solomon code (RS(204-188)). It features: high quality sound, multimedia, mobile reception, SFN's, use of the MPEG-AAC audio coding and MPEG-2 transport stream structure based on 188 bytes [ISO/IEC 13818-1].

The ISDB-T is intended for multimedia services. It basically is an improved and more flexible DVB-T system which also includes the basic features of the Eureka-147 DAB system with a large time interleaver for mobile services. It is based on band segments which are subsets of the 6 MHz NTSC channels (6/14 MHz = 430 kHz) to facilitate frequency usage and channel tuning in a 6 MHz raster. Various numbers of segments can be used in parallel along with various levels of modulation (from DQPSK for mobile reception to QPSK, 16-QAM and 64-QAM for fixed reception) to provide different channel capacity, from audio up to full HDTV services. The data throughput can vary from 280 kbit/s to 1.8 Mbit/s per 430 kHz-wide segment. In order to secure backward compatibility with simpler receivers, frequency interleaving is limited to one segment, therefore limiting the bandwidth over which short echoes can be compensated to 430 kHz. This limits the frequency diversity to one segment which is better than FM, but not as good as what it could be as seen in section 3.3.

This family of systems will result in very flexible but probably complex receivers, using basically the same technology throughout the UHF band to receive radio, TV, HDTV and datacasting services. Typically, narrowband ISDB-T will require 1 or 3 segments to transmit audio and data, whereas wideband ISDB-T carrying DTV will require 13 COFDM segments (a full 6 MHz channel).

Overall, this is an extremely flexible scheme that misses a bit on the frequency diversity advantage of wideband transmission and will require very complex receivers. It has the advantage of allowing the progressive re-farming of the UHF TV spectrum in an elegant way. So far, only Japan has officially committed to the ISDB-T system. The service is expected to be available in 2005-2007.

10.4 WorldSpace – Terrestrial augmentation

WorldSpace has proposed a terrestrial augmentation to their satellite system operating at 1.5 GHz [34]. The WorldSpace system was originally designed to provide satellite digital audio and data broadcasting for reception by outdoor fixed and portable receivers. The system was designed to optimize performance for satellite delivery using coherent QPSK modulation with block and convolutional coding, and allow operation of the satellite transponders at saturation.

The terrestrial augmentation is aimed at re-transmitting the same transport data stream as transmitted over satellite for mobile reception and to cover shadowed areas. A multi-carrier modulation (MCM) was chosen to allow operation in a multipath environment, and particularly to allow for networks of on-channel transmitters to cover cities. In fact, the system uses COFDM/DQPSK modulation. As in Eureka-147 DAB system, a guard interval is inserted between MCM symbols by time domain compressing and repeating parts of the output sequence of the IFFT. Additional sequences for framing are inserted to enhance the synchronization (timing and carrier frequency). The main system parameters are summarized in Table 8.

FFT length	512
Active carriers	480
Mapping	DQPSK (2 bit per sub-carrier) 960 bits per MCM symbol
Symbol duration	307.69 μs
Guard interval	61.54 μs
Framing	116 symbols (36 ms)
Length of synchronization preamble (inserted at the beginning of each frame)	640 samples (307.69 μsec.)
Sampling frequency	2.08 MHz
Bandwidth	1.95 MHz

Table 8: WorldSpace MCM parameters

10.5 CDM Sound BSS in Japan

Toshiba and the Association of Radio Industries and Business (ARIB) in Japan developed a system for satellite broadcasting of audio programs at 2.6 GHz based on Coded Division Multiplexing technology [35]. The same technology is expected to be used for the terrestrial augmentation through on-channel repeaters. The system will rely on RAKE techniques and antenna diversity at the receiver. The channel bandwidth is 25 MHz, The Chip rate is 16.384 MHz and the processing gain is 64. The same basic transport data stream as in ISDB is likely to be used along with a RS(204,188) block error correction coding concatenated with an inner convolutional code. This system is likely to require a rather complex receiver in order to operate in multipath environment (the Doppler spread is about 3 times that experimented in the 770 MHz band). It has similar constraints as TDM technology with respect to signal recovery in a SFN environment where equal amplitude echoes can reach the receiver. The system was field tested earlier in 1999 and the results are likely to be available soon. ARIB is planning to bring this system forward for standardization in Japan and at the ITU-R.

10.6 ATSC Terrestrial DTV Standard

The ATSC terrestrial DTV standard, using Trellis Coded 8-VSB modulation was designed for maximum data throughput over a 6 MHz channel assuming a roof-topYagi-type receiving antenna as typical reception installation.

10.7 DVB-T Terrestrial DTV Standard

The European DVB-T terrestrial broadcasting system was originally designed for fixed reception. However, because of the fact that it uses COFDM with different levels of modulation from DQPSK to 64-QAM and various sizes of guard interval, it has also been considered for mobile reception. The DQPSK and 16-QAM versions have been tested with some success in some mobile reception environment. The main drawback is that the system had not been originally designed for mobile reception and the absence of time interleaving will limit its potential in this area. The mobile reception performance of the DVB-T system might be acceptable for some mobile TV and audio applications where relatively high transmission bit error rates and intermittent reception may be acceptable but will likely be unacceptable for mobile data services where nearly errorless transmission is required for most applications.

11. Conclusions

In this report, a review of the characteristics of the transmission channel applicable to a digital broadcasting system aimed at reaching mobile receivers was made. A mobile multipath channel is a very harsh environment for a digital transmission. Three fundamental aspects that need to be considered in designing a digital transmission system for this type of channel were identified: most echoes are limited to an excess delay of about 30 μsec, the minimum channel bandwidth that is needed to provide adequate frequency diversity is about 2 MHz, and the time variability of the channel at 770 MHz for a vehicle speed of 120 km/h limits the useful symbol time to a maximum of 500 μsec.

A review of a number of modulation approaches was made which permitted to identify the strength and weaknesses of each ones in the context of a mobile multipath channel. It was found that the COFDM/DQPSK modulation is the most robust modulation that results in the simplest receiver implementation since it does not require a prediction of the channel behavior but rather a relatively simple synchronization algorithm that will make sure that most of the channel distortion can be eliminated with the receiver discarding the incoming signal during the guard interval.

Since this COFDM modulation uses echoes constructively, it brings the possibility of using on-channel repeaters to extend the coverage area and implement gap-fillers to cover shadowed areas. A host of new possibilities in implementation schemes becomes accessible to allow progressive implementation and improvement of the coverage as well as tighter control of the interfering field strength falling outside the service area.

A MMBS strawman system was developed with the characteristics listed in Table 9. The minimum field strength requirement was developed from these basic parameters: (41 dB(uV/m)). Some coverage prediction exercises over the San Francisco Bay were carried out to demonstrate the feasibility of the system and the advantage of Single-Frequency-Network.

An example of a channel allotment was developed tailored on the DTV UHF allotment plan and it was found that, typically, major cities in the US would have from 4 to 9 MMBS channels allocated if the full 24 channels that would fit in the six 6-MHz channels are available. This is based on a service availability of F(90,90) and on the use of Single Frequency Networks to minimize interference in neighboring coverage areas. Each MMBS channel can carry 3 multichannel audio/data (64kbit/s) programs.

Channel bandwidth	1.5 MHz
Modulation	COFDM/DQPSK
Channel coding	Convolutional, rate ½ [Possible puncturing from ½ to 7/8] Reed-Solomon outer code (e.g., RS(204,188)
Useful symbol period	500 μsec
Guard interval	125 μsec
Carrier spacing	2 kHz
Number of carriers	768
Distance between transmitters	SFN: 50 km, off-air pick-up re-transmitters: 25 km
Spectrum efficiency	0.75 bit/s/Hz
Capacity	1.125 Mbit/s

Table 9: Summary of the MMBS Strawman parameters

A review of the latest developments in digital broadcasting showed that COFDM is the method of choice for operation in a mobile multipath channel, proving the technical validity of this technique that was developed some 12 years ago as part of the initial design of the Eureka-147 DAB system.

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